CLAIMS

What is claimed is:

1. A method of postfiltering a speech signal using linear predictive coefficients of the speech signal for enhancing human perceptual quality of the speech signal, the method comprising the steps of:

generating a postfilter by performing a non-linear transformation of the linear predictive coefficients spectrum in the frequency domain;

applying the generated postfilter to the synthesized speech signal in the frequency domain; and

transforming the filtered frequency domain synthesized speech signal into a speech signal in the time domain.

2. The method of claim 1, wherein the step of generating a postfilter further comprises the steps of:

computing the tilt of the linear predictive coefficients spectrum in the time domain; and

compensating the linear predictive coefficients spectrum using the computed tilt in the time domain.

3. The method of claim 2, wherein the step of compensating further comprises applying a zero-padding technique.

4. The method of claim 1, wherein the step of generating a postfilter further comprises the steps of:

representing the linear predictive coefficients spectrum by a time domain vector;

transforming the time domain vector into a frequency domain vector by a Fourier transformation;

inversing the frequency domain vector; and

calculating gains according to the magnitude of the all-pole model vector, wherein the gains include a magnitude and a phase response.

5. The method of claim 4, wherein the step of calculating the gains further comprises the steps of:

normalizing the magnitude of the all-pole model vector;

conducting a non-linear transformation for the normalized magnitude of the all-pole model vector to obtain the magnitude of the gains;

estimating the phase response of the gains; and

forming the gains by combining the magnitude and the estimated phase response of the gains.

6. The method of claim 5, wherein the step of estimating the phase response further comprises executing a fast Fourier transformation based phase shifter on the gains.

7. The method of claim 1, wherein the step of generating a postfilter further comprises executing an anti-aliasing procedure in the time domain after the step of calculating the gains.

8. The method of claim 4, wherein the all-pole model is represented by a logarithm of the inverse magnitude of the frequency domain linear predictive coefficients vector.

9. The method of claim 5, wherein the non-linear transformation function comprises a scaling function with a scaling factor between 0 and 1.

10. A computer-readable medium having computer-readable instructions for performing steps to postfilter a synthesized speech signal using the linear predictive coefficients spectrum of the speech signal comprising the steps of:

computing the tilt of the linear predictive coefficients spectrum;

compensating the linear predictive coefficients spectrum using the computed tilt;

generating a postfilter by executing a non-linear transformation of the compensated linear predictive coefficients spectrum in the frequency domain; and

applying the generated postfilter to the synthesized speech signal in the frequency domain.

11. The computer-readable medium of claim 10, wherein the step of generating a postfilter further comprises the steps of:

representing the linear predictive coefficients by a time domain vector;

transforming the time domain vector into a frequency domain vector by a Fourier transformation;

transferring the frequency domain vector into an all-pole model vector; and

calculating gains according to the magnitude of the all-pole model vector, wherein the gains include a magnitude and phase response.

12. The computer-readable medium of claim 11, wherein step of calculating the gains further comprises the steps of:

normalizing the magnitude of the all-pole model vector;

conducting a non-linear transformation for the normalized magnitude of the all-pole model vector to obtain the magnitude of the gains;

estimating the phase response of the gains; and

forming the gains by combining the magnitude and the estimated phase response of the gains.

13. The computer-readable medium of claim 12, wherein the step of estimating the phase response further comprises executing a fast Fourier transformation based phase shifter.

14. The computer-readable media of claim 10, wherein the step of generating a postfilter further comprises executing an anti-aliasing procedure in the time domain.

15. The computer-readable medium of claim 11, wherein the all-pole model is represented by a logarithm of the inverse magnitude of the frequency domain vector.

16. The computer-readable media of claim 12, wherein the non-linear transformation function comprises a scaling function with a scaling factor between 0 and 1.

17. An apparatus for postfiltering a speech signal using a plurality of linear predictive coefficients of the speech signal for enhancing human perceptual quality of the speech signal, the apparatus comprising:

a Fourier transformation module operable for conducting a Fourier transformation;

an inverse Fourier transformation module operable for conducting an inverse Fourier transformation; and

a formant filter comprising formant filter gains, wherein the gains are calculated in the frequency domain by performing a non-linear transformation of the linear predictive coefficients.

18. The apparatus of claim 17, wherein the formant filter further comprises:

a linear predictive coefficients tilt computation module for computing the tilt of the linear predictive coefficients spectrum;

a linear predictive coefficients tilt compensation module for compensating the linear predictive coefficients according to the computed tilt of the linear predictive coefficients spectrum;

a formant gain calculation module for calculating formant filter gains in the frequency domain by performing a non-linear transformation of the linear predictive coefficients after tilt compensation, wherein the gains include a magnitude and phase response; and

a gain application module for applying the format filter gains to a speech signal by multiplying the gains and the speech signal in the frequency domain.

19. The apparatus of claim 18, wherein the formant gain calculation module further comprises:

a linear predictive coefficients representation module for representing the linear predictive coefficients by a time domain vector;

a modeling module for modeling a frequency domain vector according to a predefined model for generating a magnitude, wherein the frequency domain vector is transformed from the time domain vector representing the LPC coefficients;

a linear predictive coefficients non-linear transformation module for performing a non-linear transformation on the magnitude and producing the magnitude of the formant filter gains;

a phase computation module for computing a phase response of the formant filter gains according to the magnitude of the model after non-linear transformation;

a formant filter gain combination module for combining the magnitude and the phase response of the formant filter gain; and

an anti-aliasing module for preventing aliasing caused by application of the formant filter.

20. The apparatus of claim 19, wherein the linear predictive coefficients representation module is adapted for representing the linear predictive coefficients by a zero-padding technique.

21. The apparatus of claim 19, wherein the linear predictive coefficients non-linear transformation module further comprises a scaling function with a scaling factor of between 0 and 1.

22. The apparatus of claim 19, wherein the phase computation module further comprises a Hilbert phase shifter in the time domain.

23. An apparatus for use with a postfilter for processing linear predictive coefficients of a signal and providing a frequency domain formant filter gains for a formant filter, the apparatus comprising:

a linear predictive coefficients tilt computation module for computing the tilt of the linear predictive coefficients;

a linear predictive coefficients tilt compensation module for compensating the linear predictive coefficients spectrum according to the computed tilt of the linear predictive coefficients spectrum; and

a formant filter gain computation module for calculating the frequency domain formant filter gains according to the linear predictive coefficients, wherein the gains include a magnitude and a phase response.